DTMF Relay

The DTMF Relay feature allows CUBE to send dual-tone multi-frequency (DTMF) digits over IP.

This chapter talks about DTMF tones, DTMF relay mechanisms, how to configure DTMF relays, and interoperability and priority with multiple relay methods.

- Feature Information for DTMF Relay, on page 1
- Information About DTMF Relay, on page 2
- Verifying DTMF Relay, on page 10

Feature Information for DTMF Relay

The following table provides release information about the feature or features described in this module. This table lists only the software release that introduced support for a given feature in a given software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to www.cisco.com/go/cfn. An account on Cisco.com is not required.

Table 1: Feature Information for DTMF Relay

<table>
<thead>
<tr>
<th>Feature Name</th>
<th>Releases</th>
<th>Feature Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>DTMF Relay</td>
<td>Cisco IOS 12.1(2)T</td>
<td>The DTMF Relay feature allows CUBE to send DTMF digits over IP.</td>
</tr>
<tr>
<td></td>
<td>Cisco IOS XE 2.1</td>
<td>The <code>dtmf-relay</code> command was added.</td>
</tr>
<tr>
<td>Support for <code>sip-info</code> to <code>rtp-nte</code></td>
<td>Cisco IOS XE Everest 16.6.1</td>
<td>This feature adds support for <code>sip-info</code> to <code>rtp-nte</code> DTMF relay mechanism for SIP-SIP calls.</td>
</tr>
</tbody>
</table>
Information About DTMF Relay

DTMF Tones

DTMF tones are used during a call to signal to a far-end device; these signals may be for navigating a menu system, entering data, or for other types of manipulation. They are processed differently from the DTMF tones that are sent during the call setup as part of the call control. TDM interfaces on Cisco devices support DTMF by default. Cisco VoIP dial-peers do not support DTMF relay by default and require DTMF relay capabilities to be enabled.

Note

DTMF tones sent by phones do not traverse the CUBE.

DTMF Relay

Dual-tone multi-frequency (DTMF) relay is the mechanism for sending DTMF digits over IP. The VoIP dial peer can pass the DTMF digits either in band or out of band.

In-band DTMF-Relay passes the DTMF digits using the RTP media stream and uses a special payload type identifier in the RTP header to distinguish DTMF digits from actual voice communication. This method is more likely to work on lossless codecs, such as G.711.

Note

The main advantage of DTMF relay is that low bandwidth codecs like G.729 and G.723 is sent with greater fidelity when sent using in-band DTMF relay. Without the use of DTMF relay, calls established with low-bandwidth codecs may have trouble accessing automated DTMF-based systems, such as voice mail, menu-based Automatic Call Distributor (ACD) systems, and automated banking systems.

Out-of-band DTMF-Relay passes DTMF digits using a signaling protocol (SIP or H.323) instead of using the RTP media stream.

DTMF relay prevents loss of integrity of DTMF digits caused by VoIP compressed codecs. The relayed DTMF is then regenerated transparently on the peer side.

Figure 1: DTMF Relay Mechanism
DTMF relay mechanisms supported on VoIP dial-peers are listed below based on the keywords used to configure them. The DTMF relay mechanism can be either out-of-band (H.323 or SIP) or inband (RTP).

- **h245-alphanumeric and h245-signal** — These two methods are available only on H.323 dial peers. This is an out-of-band DTMF relay mechanism that transports the DTMF signals using H.245, which is the media control protocol of the H.323 protocol suite.

  The H245-signal method carries more information about the DTMF event (such as its actual duration) than the H245-Alphanumeric method. This addresses a potential problem with the alphanumeric method when interworking with other vendors’ systems.

- **sip-notify** — This method is available on SIP dial peers only. This is a Cisco proprietary out-of-band DTMF relay mechanism that transports DTMF signals using SIP-Notify message. The SIP Call-Info header is used to indicate the use of the SIP-Notify DTMF relay mechanism. The message is acknowledged with a 18x or 200 response message containing a similar SIP Call-Info header.

  The Call-Info header for NOTIFY-based out-of-band relay is as follows:

  ```
  Call-Info: <sip: address>; method="NOTIFY;Event=telephone-event;Duration=msec"
  ```

  DTMF relay digits are sent as 4 bytes in a binary encoded format.

  This mechanism is useful for communicating with SCCP IP phones that do not support inband DTMF digits and analog phones attached to analog voice ports (FXS) on the router.

  If multiple DTMF relay mechanisms are enabled on a SIP dial peer and are negotiated successfully, NOTIFY-based out-of-band DTMF relay takes precedence.

- **sip-kpml** — This method is available only on SIP dial peers. This is an out-of-band DTMF relay mechanism defined by RFC 4730 that registers the DTMF signals using SIP-Subscribe messages and transports the DTMF signals using SIP-Notify messages containing an XML-encoded body. This method is also known as Key Press Markup Language.

  If you configure KPML on the dial peer, the gateway sends INVITE messages with kpml in the Allow-Events header.

  This method is mostly used for SIP endpoints registered to CUCM or CME. This method is useful for non-conferencing calls and for interoperability between SIP products and SIP phones.

  If you configure rtp-nte, sip-notify, and sip-kpml, the outgoing INVITE contains an SDP with rtp-nte payload, a SIP Call-Info header, and an Allow-Events header with KPML.

  The following SIP-Notify message is a sample taken after the subscription has taken place. The endpoints transmit digits using SIP-Notify messages with KPML events through XML. In the following example, the digit “1” is being transmitted:

  ```
  NOTIFY sip:192.168.105.255:5060 SIP/2.0
  Event: kpml
  <?xml version="1.0" encoding="UTF-8"?>
  <kpml-response version="1.0" code="200" text="OK" digits="1" tag="dtmf"/>
  ```

- **sip-info** — The sip-info method is available only on SIP dial peers. This is an out-of-band DTMF relay mechanism that registers the DTMF signals using SIP-Info messages. The body of the SIP message consists of signaling information and uses the content-type application/dtmf-relay.

  The method is always enabled for SIP dial peers, and is invoked when a SIP INFO message is received with DTMF relay content.

  This following sample message shows a SIP INFO message received by the gateway with specifics about the DTMF tone to be generated. The combination of the From, To, and Call-ID headers identifies the
call leg. The signal and duration headers specify the digit, in this case 1, and duration, 160 milliseconds in the example, for DTMF tone play.

INFO sip:2143302100@172.17.2.33 SIP/2.0
Via: SIP/2.0/UDP 172.180.2.100:5060
From: <sip:9724401003@172.180.2.100>;tag=43
To: <sip:2143302100@172.17.2.33>;tag=9753.0207
Call-ID: 984072_15401962@172.80.2.100
CSeq: 25634 INFO
Supported: 100rel
Supported: timer
Content-Length: 26
Content-Type: application/dtmf-relay
Signal= 1
Duration= 160

• rtp-nent—Real-Time Transport Protocol (RTP) Named Telephone Events (NTE). This is an in-band DTMF relay mechanism defined by RFC2833. RFC2833 defines formats of NTE-RTP packets used to transport DTMF digits, hook flash, and other telephony events between two peer endpoints. DTMF tones are sent as packet data after call media has been established using the RTP stream and are distinguished from the audio by the RTP payload type field, preventing compression of DTMF-based RTP packets. For example, the audio of a call can be sent on a session with an RTP payload type that identifies it as G.711 data, and the DTMF packets are sent with an RTP payload type that identifies them as NTEs. The consumer of the stream utilizes the G.711 packets and the NTE packets separately.

The SIP NTE DTMF relay feature provides reliable digit relay between Cisco VoIP gateways when a low-bandwidth codec is used.

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**Note**

Payload type 96 and 97 are used for fax by default in Cisco devices. A third party device may use payload type 96 and 97 for dtmf. In such scenarios, we recommend you to perform one of the following:

• Change the payload type for fax in both incoming and outgoing dial-peers using `rtp payload-type` command

• Use `assymmetric payload dtmf` command

For more information on configuring rtp payload-type and assymmetric payload dtmf, see [Dynamic Payload Type Interworking for DTMF and Codec Packets for SIP-to-SIP Calls](#).

Payload types and attributes of this method are negotiated between the two ends at call setup using the Session Description Protocol (SDP) within the body section of the SIP message.

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**Note**

This method should not be confused with the “Voice in-band audio/G711” transport because the latter is just the audible tones being passed as normal audio without any relay signaling method being “aware” or involved in the process. This is just plain audio passing through end-to-end using the G711Ulaw/Alaw codec.

• cisco-rtp—This is an in-band DTMF relay mechanism that is Cisco proprietary, where the DTMF digits are encoded differently from the audio and are identified as payload type 121. The DTMF digits are part
of the RTP data stream and distinguished from the audio by the RTP payload type field. This method is not supported by CUCM and its use has been discontinued.

Note

The cisco-rtp operates only between two Cisco 2600 series or Cisco 3600 series devices. Otherwise, the DTMF relay feature does not function, and the gateway sends DTMF tones in-band.

- **G711 audio**—This is an inband DTMF relay mechanism that is enabled by default and requires no configuration. Digits are transmitted within the audio of the phone conversation, that is, it is audible to the conversation partners; therefore, only uncompressed codecs like g711 alaw or ulaw can carry inband DTMF reliably. Female voices are known to, sometimes, trigger the recognition of a DTMF tone. Digits are passed along just like the rest of your voice as normal audio tones with no special coding or markers using the same codec as your voice does and are generated by your phone.

### Configuring DTMF Relays

You can configure DTMF relay using the `dtmf-relay method1 [..[method6]]` command in the VoIP dial peer. DTMF negotiation is performed based on the matching inbound dial-peer configuration.

The `method` variable used here can be any of the following:

- h245-alphanumeric
- h245-signal
- sip-notify
- sip-kpml
- sip-info
- rtp-nte [digit-drop]
- cisco-rtp

Multiple DTMF methods may be configured on CUBE simultaneously in order to minimize MTP requirements. If you configure more than one out-of-band DTMF method, preference goes from highest to lowest in the order of configuration. If an endpoint does not support any of the DTMF relay mechanism configured on CUBE, an MTP or transcoder is required.

The following table lists the DTMF relay types supported on a SIP and H.322 gateway.

<table>
<thead>
<tr>
<th>Gateway Type</th>
<th>H.323 Gateway</th>
<th>SIP Gateway</th>
</tr>
</thead>
<tbody>
<tr>
<td>In-band</td>
<td>cisco-rtp, rtp-nte</td>
<td>rtp-nte</td>
</tr>
<tr>
<td>Out-of-band</td>
<td>h245-alphanumeric, h245-signal</td>
<td>sip-notify, sip-kpml, sip-info</td>
</tr>
</tbody>
</table>

### Interoperability and Priority with Multiple DTMF Relay Methods

- CUBE negotiates both rtp-nte and sip-kpml if both are supported and advertised in the incoming INVITE. However, CUBE relies on the rtp-nte DTMF method to receive digits and a SUBSCRIBE if sip-kpml
is not initiated. CUBE still accepts SUBSCRIBEs for KPML. This prevents double-digit reporting problems at CUBE.

- CUBE negotiates to one of the following:
  - cisco-rtp
  - rtp-nte
  - rtp-nte and kpml
  - kpml
  - sip-notify

- If you configure rtp-nte, sip-notify, and sip-kpml, the outgoing INVITE contains a SIP Call-Info header, an Allow-Events header with KPML, and an sd with rtp-nte payload.

- If you configure more than one out-of-band DTMF method, preference goes from highest to lowest in the order of configuration.

- CUBE selects DTMF relay mechanisms using the following priority:
  - sip-notify or sip-kpml (highest priority)
  - rtp-nte
  - None—Send DTMF in-band

H.323 gateways select DTMF relay mechanisms using the following priority:

- cisco-rtp
- h245-signal
- h245-alphanumeric
- rtp-nte
- None—Send DTMF in-band

**DTMF Interoperability Table**

This table provides the DTMF interoperability information between various DTMF relay types in different call flow scenarios. For instance, if you need to configure sip-kpml on an inbound dial peer and h245-signaling on an outbound dial peer in an RTP-RTP Flow through configuration, refer table 1 to see that the combination is supported (as image information is present) and the required image is IOS 12.4(15)T or IOS XE Supported or above. The call scenarios provided are as follows:

- RTP-RTP Flow-Through
- RTP-RTP with transcoder Flow-Through
- RTP-RTP Flow Around
- RTP-RTP with high-density transcoder Flow Through
- SRTP-RTP Flow Through
### Table 3: RTP-RTP Flow-Through

<table>
<thead>
<tr>
<th>dial-peer protocol</th>
<th>DTMF Relay Type</th>
<th>H.323</th>
<th>SIP</th>
<th>Inband</th>
</tr>
</thead>
<tbody>
<tr>
<td>H.323</td>
<td>h245-alpha numeric</td>
<td>Supported</td>
<td>Supported</td>
<td>Supported</td>
</tr>
<tr>
<td></td>
<td>h245-signal</td>
<td>Supported</td>
<td>Supported</td>
<td>Supported</td>
</tr>
<tr>
<td></td>
<td>rtp-nte</td>
<td>Supported</td>
<td>Supported</td>
<td>Supported</td>
</tr>
<tr>
<td></td>
<td>sip-kpml</td>
<td>Supported</td>
<td>Supported</td>
<td>Supported</td>
</tr>
<tr>
<td></td>
<td>sip-notify</td>
<td>Supported</td>
<td>Supported</td>
<td>Supported</td>
</tr>
<tr>
<td></td>
<td>sip-info</td>
<td>Supported</td>
<td></td>
<td>Supported†</td>
</tr>
<tr>
<td>Inband</td>
<td>Voice Inband (G.711)</td>
<td>Supported*</td>
<td>Supported*</td>
<td>Support†</td>
</tr>
<tr>
<td>dial-peer protocol</td>
<td>H.323</td>
<td>SIP</td>
<td>Inband</td>
<td></td>
</tr>
<tr>
<td>--------------------</td>
<td>-------</td>
<td>-----</td>
<td>--------</td>
<td></td>
</tr>
<tr>
<td>Inband</td>
<td></td>
<td></td>
<td>Voice Inband (G.711)</td>
<td></td>
</tr>
</tbody>
</table>

**Table 5: RTP-RTP Flow Around**

<table>
<thead>
<tr>
<th>dial-peer protocol</th>
<th>H.323</th>
<th>SIP</th>
<th>Inband</th>
</tr>
</thead>
<tbody>
<tr>
<td>H.323</td>
<td>h245-alpha numeric</td>
<td>rtp-n-te</td>
<td>Supported*</td>
</tr>
<tr>
<td></td>
<td>h245-signal</td>
<td>sip-kpml</td>
<td>Supported*</td>
</tr>
<tr>
<td></td>
<td>rtp-n-te</td>
<td>sip-notify</td>
<td>Supported</td>
</tr>
<tr>
<td>SIP</td>
<td>rtp-n-te</td>
<td>sip-kpml</td>
<td>Supported*</td>
</tr>
<tr>
<td></td>
<td>sip-notify</td>
<td>sip-info</td>
<td>Supported*</td>
</tr>
<tr>
<td>Inband</td>
<td>Voice Inband (G.711)</td>
<td>sip-info</td>
<td>Supported</td>
</tr>
</tbody>
</table>

* media resource is required (Transcoder) for IOS versions. CUBE falls back to flow-through mode if media resource is unavailable.
### Table 6: RTP-RTP with high-density transcoder Flow Through

<table>
<thead>
<tr>
<th>dial-peer protocol</th>
<th>H.323</th>
<th>SIP</th>
<th>Inband</th>
</tr>
</thead>
<tbody>
<tr>
<td>dial-peer protocol</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>DTMF Relay Type</td>
<td>h245-alpha numeric</td>
<td>h245-signal</td>
<td>rtp-nte</td>
</tr>
<tr>
<td>H.323</td>
<td>Supported</td>
<td></td>
<td></td>
</tr>
<tr>
<td>h245-signal</td>
<td>Supported</td>
<td></td>
<td></td>
</tr>
<tr>
<td>rtp-nte</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>SIP</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>rtp-nte</td>
<td>Supported</td>
<td>Supported</td>
<td>Supported</td>
</tr>
<tr>
<td>sip-kpml</td>
<td>Supported</td>
<td>Supported</td>
<td></td>
</tr>
<tr>
<td>sip-notify</td>
<td>Supported</td>
<td></td>
<td></td>
</tr>
<tr>
<td>sip-info</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Inband</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Voice Inband (G.711)</td>
<td>Supported</td>
<td>Supported</td>
<td></td>
</tr>
</tbody>
</table>

### Table 7: SRTP-RTP Flow Through

<table>
<thead>
<tr>
<th>dial-peer protocol</th>
<th>H.323</th>
<th>SIP</th>
<th>Inband</th>
</tr>
</thead>
<tbody>
<tr>
<td>dial-peer protocol</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>DTMF Relay Type</td>
<td>H.323</td>
<td>h245-alpha numeric</td>
<td>h245-signal</td>
</tr>
<tr>
<td>H.323</td>
<td></td>
<td>Supported</td>
<td></td>
</tr>
<tr>
<td>h245-signal</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>rtp-nlte</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>SIP</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>rtp-nlte</td>
<td></td>
<td>Supported</td>
<td></td>
</tr>
<tr>
<td>sip-kpml</td>
<td></td>
<td>Supported</td>
<td></td>
</tr>
<tr>
<td>sip-notify</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>sip-info</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Inband</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Voice Inband (G.711)</td>
<td></td>
<td>Supported</td>
<td></td>
</tr>
</tbody>
</table>
Verifying DTMF Relay

SUMMARY STEPS

1. show sip-ua calls
2. show sip-ua calls dtmf-relay sip-info
3. show sip-ua history dtmf-relay kpml
4. show sip-ua history dtmf-relay sip-notify

DETAIL STEPS

Step 1 show sip-ua calls

The following sample output shows that the DTMF method is SIP-KPML.

Example:

Device# show sip-ua calls

SIP UAC CALL INFO
Call 1
SIP Call ID : 57633F68-2BE011D6-8013D46B-B4F9B5F6@172.18.193.251
State of the call : STATE_ACTIVE (7)
Substate of the call : SUBSTATE_NONE (0)
Calling Number : 
Called Number : 8888
Bit Flags : 0xD44018 0x100 0x0
CC Call ID : 6
Source IP Address (Sig ) : 192.0.2.1
Destn SIP Req Addr:Port : 192.0.2.2:5060
Destn SIP Resp Addr:Port: 192.0.2.3:5060
Destination Name : 192.0.2.4.250
Number of Media Streams : 1
Number of Active Streams: 1
RTP Fork Object : 0x0
Media Mode : flow-through
Media Stream 1
State of the stream : STREAM_ACTIVE
Stream Call ID : 6
Stream Type : voice-only (0)
Negotiated Codec : g711ulaw (160 bytes)
Codec Payload Type : 0
Negotiated Dtmf-relay : sip-kpml
Dtmf-relay Payload Type : 0
Media Source IP Addr:Port: 192.0.2.5:17576
Media Dest IP Addr:Port : 192.0.2.6:17468
Step 2 show sip-ua calls dtmf-relay sip-info

The following sample output displays active SIP calls with INFO DTMF Relay mode.

Example:

Device# show sip-ua calls dtmf-relay sip-info

Total SIP call legs:2, User Agent Client:1, User Agent Server:1

SIP UAC CALL INFO
Call 1
SIP Call ID : 9598A547-5C1311E2-8008F709-2470C996@172.27.161.122
State of the call : STATE_ACTIVE (?)
Calling Number : sipp
Called Number : 3269011111
CC Call ID : 2
No. Timestamp Digit Duration
0 01/12/2013 17:23:25.615 2 250
1 01/12/2013 17:23:25.967 5 300
2 01/12/2013 17:23:26.367 6 300

Call 2
SIP Call ID : 1-29452@172.25.208.177
State of the call : STATE_ACTIVE (?)
Calling Number : sipp
Called Number : 3269011111
CC Call ID : 1
No. Timestamp Digit Duration
0 01/12/2013 17:23:25.615 2 250
1 01/12/2013 17:23:25.967 5 300
2 01/12/2013 17:23:26.367 6 300

Number of SIP User Agent Client(UAC) calls: 2

SIP UAS CALL INFO
Call 1
SIP Call ID : 1-29452@172.25.208.177
State of the call : STATE_ACTIVE (?)
Calling Number : sipp
Called Number : 3269011111
CC Call ID : 3
No. Timestamp Digit Duration
0 01/12/2013 17:23:25.615 2 250
1 01/12/2013 17:23:25.967 5 300
2 01/12/2013 17:23:26.367 6 300

Call 2
SIP Call ID : 9598A547-5C1311E2-8008F709-2470C996@172.27.161.122
State of the call : STATE_ACTIVE (?)
Calling Number : sipp
Called Number : 3269011111
CC Call ID : 2
No. Timestamp Digit Duration
0 01/12/2013 17:23:25.615 2 250
1 01/12/2013 17:23:25.967 5 300
Step 3  show sip-ua history dtmf-relay kpml

The following sample output displays SIP call history with KMPL DTMF Relay mode.

Example:

Device# show sip-ua history dtmf-relay kpml

Total SIP call legs:2, User Agent Client:1, User Agent Server:1
SIP UAC CALL INFO
Call 1
SIP Call ID : D0498774-F01311E3-82A0DE9F-78C438FF010.86.176.119
  State of the call : STATE_ACTIVE (7)
  Calling Number : 2017
  Called Number : 1011
  CC Call ID : 257
No.  Timestamp  Digit  Duration
-------------------------------------------------------------------------------------
Call 2
SIP Call ID : 22BC36A5-F01411E3-81808A6A-5FE95113010.86.176.142
  State of the call : STATE_ACTIVE (7)
  Calling Number : 2017
  Called Number : 1011
  CC Call ID : 256
No.  Timestamp  Digit  Duration
-------------------------------------------------------------------------------------
Number of SIP User Agent Client(UAC) calls: 2

SIP UAS CALL INFO
Call 1
SIP Call ID : 22BC36A5-F01411E3-81808A6A-5FE95113010.86.176.142
  State of the call : STATE_ACTIVE (7)
  Calling Number : 2017
  Called Number : 1011
  CC Call ID : 256
No.  Timestamp  Digit  Duration
-------------------------------------------------------------------------------------
Call 2
SIP Call ID : D0498774-F01311E3-82A0DE9F-78C438FF010.86.176.119
  State of the call : STATE_ACTIVE (7)
  Calling Number : 2017
  Called Number : 1011
  CC Call ID : 257
No.  Timestamp  Digit  Duration
-------------------------------------------------------------------------------------
Number of SIP User Agent Server(UAS) calls: 2

Step 4  show sip-ua history dtmf-relay sip-notify

The following sample output displays SIP call history with SIP Notify DTMF Relay mode.

Example:

Device# show sip-ua history dtmf-relay sip-notify
Total SIP call legs: 2, User Agent Client: 1, User Agent Server: 1

SIP UAC CALL INFO

Call 1
SIP Call ID : 29BB98C-F01311E3-8297DE9F-78C438FF@10.86.176.119
  State of the call : STATE_ACTIVE (7)
  Calling Number : 2017
  Called Number : 1011
  CC Call ID : 252

Number of SIP User Agent Client(UAC) calls: 2

SIP UAS CALL INFO

Call 1
SIP Call ID : 550E973B-F01311E3-817A8A6A-5FE95113@10.86.176.142
  State of the call : STATE_ACTIVE (7)
  Calling Number : 2017
  Called Number : 1011
  CC Call ID : 251

Number of SIP User Agent Server(UAS) calls: 2
Verifying DTMF Relay